ESE 150: Digital Audio Basics — Spring 2017

Dr. Thomas Farmer — **Final Exam**

**Part I - Basic Knowledge Questions (3 points each)**

***Directions: Provide as brief an answer as possible. Record your answer in the provided white book. If I cannot read your handwriting, you are less likely to receive partial credit.***

1. In lab, you used the Arudino to sample music; it returned values from 0-1023. Are those numbers an example of sampling or quantization?

2. The following Arduino code helped you obtain music samples:

   ```
   for(int i = 0; i < 800; i++) {
      incomingAudio[i] = analogRead(A0) ;
      delayMicroseconds(300) ;
   }
   ```
   Assuming analogRead() has no delay, what is the "rate" in Hz for the samples you obtained?

3. What is the advantage/disadvantage of lossy compression compared to lossless compression?

4. What is the purpose of a "transform"

5. Why is the frequency domain representation of digital audio necessary in the MP3 algorithm; what does it enable that the time domain representation would not?

6. List the basic steps involved in the MP3 algorithm.

7. List the types of compression/decompression used in the MP3 algorithm; identify the types of each compression (lossy/lossless).

8. What does the Fourier series allow us to do for any periodic signal?

9. What is Boolean Algebra?

10. Identify a major component of the CPU that uses combinational logic and a second major component that uses sequential logic.

11. Within the context of an Operating System, what is a PCB and what is it used for?

12. What is the part of the ear that directly interprets sound?

13. In the DFT labs, why do we plot the real component of the FFT, but modify the complex output of the FFT?

**Part II – True/False, Multiple Choice, Fill in the Blank (2 points each)**

***Directions: For multiple choice problems circle ONLY one answer (unless otherwise indicated). Write your answer directly on the exam (not in the white book)***

1) How would you describe the signal that your Arduino captured in lab?
    a. Independent time variable, dependent frequency variable
    b. Independent voltage variable, dependent frequency variable
    c. Independent frequency variable, dependent voltage variable
    d. Independent time variable, dependent voltage variable

2) Huffman Code is (circle all that apply):
    a. Is a lossy form of compression
    b. Is a lossless form of compression
    c. Is a fixed length encoding algorithm
    d. Is a variable length encoding algorithm
    e. All of the above

3) Digitization refers to the following:
    a. Transforming analog signal into a binary representation
    b. Transforming continuous signal into a digital signal
    c. Using discrete values to represent an analog waveform
    d. None of the above
    e. All of the above

4) In psychoacoustic compression, the value used to directly determine bit allocation is
    a. the sound pressure level
    b. the signal to mask ratio
    c. the Schroeder equation
    d. the bark frequency
    e. the Fletcher curve

5) Which of the following are true for this R-S latch (circle all that apply):
    a. has only one stable state
    b. is an example of a combinational logic circuit
    c. must have both inputs set to 0 to store data
    d. can store, or “latch” onto, its last output
    e. is made from two NAND gates

6) The following code multiples 3 times 4. Where in the Von Neumann architecture does the bolded line occur ***(circle all that apply)?***

```
A=3, B=4, C=0 ;
while (B>0) {
   C=C+A ;
   B=B-1 ;
}
```

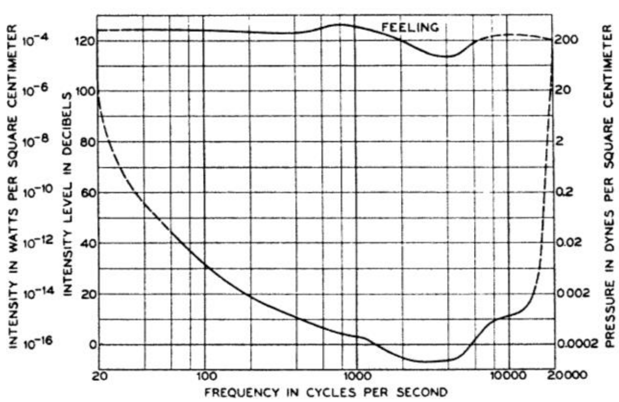
a. Output
b. Control Unit
c. ALU
d. Memory
e. PC
f. Registers
g. Input

7) (True/False) Compressibility is lower as data becomes less structured

8) (True/False) An Arduino and computer speakers are both examples of things that we use as ADC

9) (True/False) In a typical filesystem, when a file is deleted, the physically stored bits on the hard drive are permanently deleted

10) While human hearing is limited to 20 kHz, the phone company inserts a filter before their ADCs, that cuts off frequencies above _____ Hz.




a) 1000 Hz
b) 4000 Hz
c) 8000 Hz
d) 12000 Hz
e) 20000 Hz

11) Why does the phone company insert a filter before their ADC (circle all that apply)?
a. To limit unwanted noise in their system
b. Increase the signal to noise (SNR) ratio
c. As a form of compression
d. To reduce the amount of data they must transmit across their network
e. All of the above

12) When recording audio an analog filter with an upper bound of ~20kHz is inserted before the ADC to prevent aliasing. In a purely digital recording environment, an alternative to the analog filter is:
    a. Antialiasing filter
    b. A digital filter
    c. Oversampling
    d. Undersampling
    e. None of the above

13) The purpose of employing psychoacoustics in digital audio is:
    a. To improve audio clarity
    b. To decrease filesize
    c. To simplify compression
    d. All of the above
    e. None of the above

14) In a certain recording studio, the analog filter prior to the ADC has malfunctioned and is not working. A vibration in the studio at 28 kHz has produced an audible noise in the recoded music at which frequency:
    a. 28 kHz
    b. 16 kHz
    c. 8 kHz
    d. 4 kHz
    e. There will be no noise as 28 kHz is beyond the limit of human hearing

15) In a DFT, the coefficients of the cosine function represent the _____________ of the frequency components of a time domain signal, while the coefficients of the sine function represent the ____________ of the frequency components of a time signal.

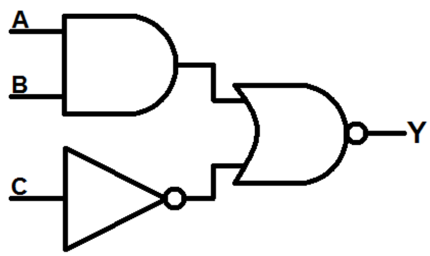
**Part III – Calculations**

***Directions: For each problem, record your answer in the provided white book. Show all your steps and circle final answers.***

1) (2 points) Convert the binary number: 011001 to decimal _________________

2) (2 points) Convert the decimal number: 45 to binary __________

3) (4 points) Given the following combinational logic circuit:
   a. Generate its truth table




   b. **e.c:** Use Boolean Algebra to simplify the above circuit to use fewer gates

4) (4 points) Given the samples: {0, 500, 1000, 700, 200, 0} obtained from an Arduino using the sketch listed in Part I, question #2, reconstruct the original signal on an x-y coordinate system. Label each axis, indicate the time (and units) and voltage (and units). Extrapolate between the data points using a straight-line approximation.

5) (6 points) Given the following dataset: {45, 96, 12, 96, 13, 45, 45, 12}:
   a. Generate the Huffman encoding Tree
   b. Make a table that show the binary encoding for each symbol in the dataset
   c. If the dataset was originally encoded in 8-bit ASCII, what would the compression ratio be when your Huffman tree is used to encode the data (use definition: bits in/bits out)

6) (4 points) Given that the lowest limit of human ear sensitivity is 20$\mu$Pascals, if the sound intensity of normal conversation is 60dB, how many dB would represent twice the sound pressure?

$$L_{SPL} = 20 * log_{10}\left(\frac{pressure}{Reference\ pressure}\right)$$

7) (9 points) You are hired as a systems engineer to architect a cloud based sound music service. Answer the following capacity planning questions.
   a. Your boss wishes to use 32-bit resolution and a 96kHz sampling frequency for audio files; how many bytes would be required to store one 3 minute stereo song (aka 2-channels)?
   b. If you wanted to store 100,000 3-minute songs, what would the capacity of the hard drive need to be for the server?
   c. How fast a connection to the internet would be required for 1000 simultaneous users to be able to download music at the same time?